Grandstream Networks, Inc.

Analog IP Gateway GXW410x
4 or 8 FXO Ports w/Video Surveillance

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GXW410X USER MANUAL

http://www.grandstream.com/support/gxw_series/gxw410x/documents/gxw410x_gui.zip

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WELCOME

Thank you for purchasing the Grandstream GXW410x IP Analog FXO Gateway. The GXW410x is a cost effective, easy to use and easy to configure IP communications solution for any business. The GXW410x gateway supports popular voice codecs and is designed for full SIP compatibility and interoperability with party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GXW FXO Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, etc. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise. Enable the video surveillance port to give you peace of mind while you are away from your business.

GATEWAY GXW410X OVERVIEW

The GXW410x offers an easy to manage, feature-rich IP communications solution for any small business or businesses with virtual and/or branch locations who want to leverage their broadband network and/or add new IP Technology to their current phone system. The Grandstream enterprise analog VoIP Gateway GXW410x series converts SIP/RTP IP calls to traditional PSTN calls and vice versa. There are two models - the GXW4104 and GXW4108, which feature 4 or 8 FXO ports respectively. The installation is the same for either model.

SAFETY COMPLIANCES

The GXW410x is compliant with various safety standards including FCC/CE. Its power adapter is compliant with UL standard. ***Warning:*** use only the power adapter included in the GXW410x package. Using an alternative power adapter may permanently damage the unit.

WARRANTY

Grandstream has a reseller agreement with our reseller customer. End-users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number. Grandstream reserves the right to remedy warranty policy without prior notification.

Caution: *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.*

- This document contains links to Grandstream GUI Interfaces. Please download the GUI examples http://www.grandstream.com/support/gxw_series/gxw410x/documents/gxw410x_gui.zip for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available on our website: http://www.grandstream.com/support/gxw_series/gxw410x/documents/gxw410x_usermanual_english.pdf

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PACKAGING

Unpack and check all accessories. Equipment included in the package:

1) One GXW410x Unit
2) One universal power adapter
3) One Ethernet cable

CONNECTING THE GXW410X

FIGURE 1: DIAGRAM OF GXW410X BACK PANEL

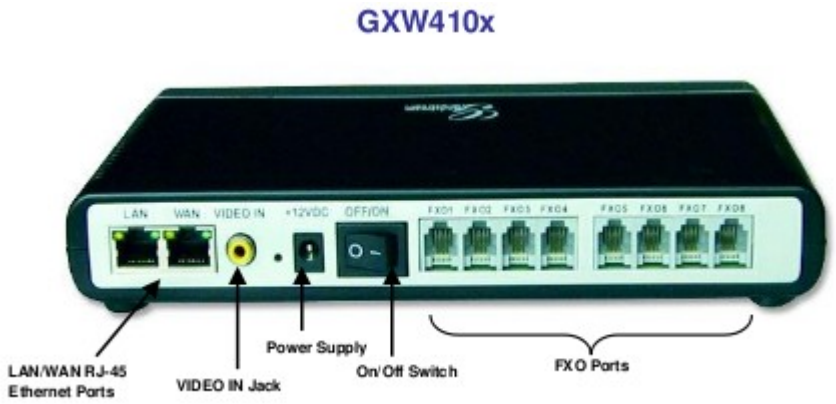


TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

LAN (or PC)	Connect your PC to this port. It will then be assigned an IP address from your Router/DHCP Server. The GXW410x acts as a switch only.
WAN	Connect to the internal LAN network or Public Internet.
VIDEO IN	Connection for Analog based Video Surveillance Camera (RCA)
RESET	Factory Reset button. Press for 7 seconds to reset factory default settings.
POWER IN	Power adapter connection
OFF/ON	Off/On switch
FXO1 - FXO8	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

Figure 2: Diagram of GXW410x Display Panel

Table 2: Definitions of the GXW Display Panel

Power LED	Indicates Power. Remains ON when Power is connected and unit is turned ON.
Ready LED	Remains ON after boot-up.
LAN LED	Indicates LAN (or WAN) port activity
PC LED	Indicates PC (or LAN) port activity
Video LED	Remains solid green on boot-up. If Video IN terminal is connected, indicates video activity.
LEDs 1 - 8	Indicate status of the respective FXO Ports on the back panel **Busy** - ON **Available** - OFF

NOTE: All LEDs display green when ON. The **Ready** light will only be ON when the network interface is ready and the Web User Interface is accessible.

During a *firmware upgrade* or *configuration download* the following LED pattern will be observed:

Power, Ready, Video and **WAN** LEDs will be **ON.** The **FXO port** LED will keep flashing during download and then stay OFF while the new files are written. The entire process may take between 20 to 30 minutes. The firmware upgrade is complete when you can login into the web configuration pages.

APPLICATION DESCRIPTION

A. IP PBX / SIP Server with GXW410x

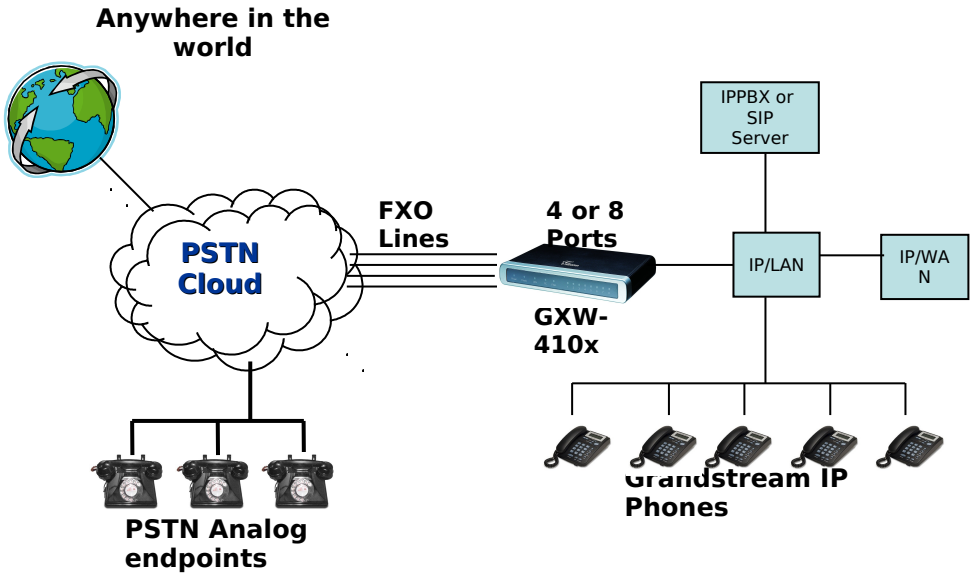
A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW410x series gateway. In this environment, the SIP server handles SIP registration and call control and the GXW410x processes media conversion between IP and PSTN calls.

There are 2 ways to configure GXW410x when using it with a SIP Server:

1. **With SIP accounts configured on Channels page.** In this case, the GXW410x gateway acts as an endpoint requesting registration from the SIP Server. Under the Channels page you will need to fill in the information like SIP User ID, Password, etc. Now, when calls are made from an IP phone, the call will be routed to the SIP Server which will forward it to one of the SIP accounts on the GXW410x, which will then forward it to the PSTN line.

2. **Without SIP accounts.** In this case, you simply have to configure the SIP Server to perform forwarding of the SIP INVITE message with the FXO destination number to the gateway's IP Address. The GXW410x gateway will receive the digits and immediately forward them on the FXO lines to the destination PSTN. Most of the configuration on the Gateway for this case will remain default, except Stage Method needs to be set to 1, and SIP Server IP Address/DNS name has to be filled.

FUNCTIONAL DIAGRAM OF IP-PBX & GXW410X

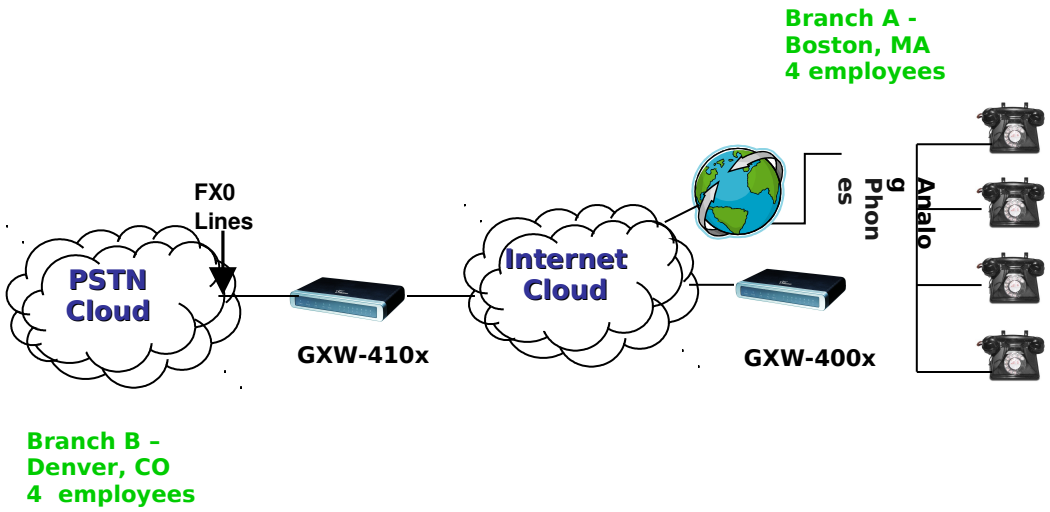


For incoming calls from the PSTN analog endpoints to the GXW410x gateway, the device will auto forward each call to a configured IP extension. The SIP Server can then route the call based on its own configuration or IVR system.

B. FXS Gateway with GXW410x [No SIP Server required]

Alternatively, the GXW410x gateway can be used without a SIP Server. It can be used in conjunction with an FXS Gateway (Ex. GXW400x) and calls can still be originated from the IP network and terminated on the PSTN network and vice versa. All you need to make sure is that both gateways are able to locate each other (i.e. they should be on the same LAN or have public IP addresses).

GXW400x & GXW410x Scenario/Toll- Free Calling Between Locations



If you setup resembles the one in this diagram, you need to configure the SIP Server field to be the IP address of the other gateway (i.e. configure the IP address of GXW400x gateway to be the SIP Server for the GXW410x gateway and vice versa). Please be sure you set SIP Registration to No.

Expected Call Flow: An analog Phone (connected to the GXW400x gateway) picks up and dials the destination PSTN number. The call gets routed to the GXW410x gateway which dials out the digit string onto the FXO Lines, thus reaching the destination PSTN endpoint. On the reverse, incoming calls from the PSTN endpoints connected to the GXW410x gateway will be routed automatically to the analog endpoints connected to the GXW400x gateway.

FXS and FXO Gateway Configuration Example

GXW400x Gateway	GXW410x Gateway
Profile 1 SIP Server - Set it to IP Address of GXW410x SIP Registration - No Outgoing Call without Registration - Yes NAT traversal – No	***Advanced Settings*** STUN Server - Blank Use Random Port - No
Advanced Settings STUN Server - Blank	***FXO lines*** Wait for Dial Tone - Y or N (whichever works for your PSTN Service Provider) Stage Method - 1 Unconditional Call Forward to VOIP: ch1-8:444; @ch1-8:p1; ch1-8:5060++;
	Channels 1-8 5060 Profile 1 Local SIP Listen port (For VOIP to PSTN calls) - 5060++
	Profile 1 SIP Server - Set it to IP Address of GXW400x SIP Registration - No NAT traversal - No

For more information regarding this setup, email Grandstream technical support at support@grandstream.com or visit our User and Developer forum at forums.grandstream.com

FEATURES

The GXW410x is a next generation IP voice and video gateway that features full interoperability with leading IP-PBXs, Soft-switches and SIP platforms. The GXW410x series gateways offer superb voice and video quality, traditional telephony functionality, simple configuration, feature-rich functionality and an additional video port that enables the gateway to act as a video surveillance gateway.

SOFTWARE FEATURES OVERVIEW

- 4 and 8 FXO port media gateways
- Video surveillance port
- External power supply
- Two RJ-45 ports (switched or routed)
- TFTP and HTTP firmware upgrade support
- Multiple SIP accounts, multiple SIP profiles (choice of 3 profiles per account)
- Supports Audio Codecs: G711U/A, G723, G729A/B and GSM
- Supports Video Codecs: H.264
- G.168 – echo cancellation
- Flexible DTMF transmission: In Audio, RFC2833, SIP Info or any combination of the 3
- Selectable, multiple LBR coders per channel
- T.38 compliant

HARDWARE SPECIFICATION

TABLE 4: HARDWARE SPECIFICATION OF GXW410X

LAN interface	2xRJ45 10/100Mbps
LED	8 LEDs (GREEN)
Universal Switching Power Adapter	**Input**: 100-240V AC, 50/60Hz, 0.5A Max **Output**: 12V DC, 1.25A **UL** certified
Dimension	225mm (L) x 172mm (W) x 42mm (H)
Weight	0.29 lbs (3.5 oz)
Temperature	32~104°F 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

Table 3: GXW410x Software Features

	GXW410x FX0 Analog Gateway Series
IP settings	GXW4104: 4 ports; 4 SIP accounts w/ choice of 3 SIP Server profiles
	GXW4108: 8 ports; 8 SIP accounts w/ choice of 3 SIP Server profiles
	Round-robin port scheduling to ensure available lines to access PSTN networks
Telephone Interface	FXO, RJ11
Network Interface	Two (2) 10/100 Mbps, RJ45
LED Indicators	Power, Video, and Line LEDs
On/Off Switch	Yes
Voice over Packet Capabilities	G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice Compression	G.711U, G711A, G.723, G.729A/B, GSM
Video Surveillance	Real-time H.264 base CIF resolution
DHCP Server/Client	Switch Mode and PPPoE
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP and RTSP
PSTN Signaling	FXO Loop start, Current Disconnect.
DTMF Method	Flexible DTMF transmission method, User interface of In-audio, RFC2833, and SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP and HTTP
Media	SRTP
Control	TLS and SIPS (pending)
Management	Syslog support, HTTPS and telnet (pending), remote management using Web browser
Short and long haul	REN3: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes (Detection only). The PSTN lines will need to be subscribed to PR service from the Service Provider.
EMC	GXW410x: EN55022 Class B, CFR Part 15 Class B, EN55024; GXW4104: FCC, CE (in addition)
Safety	GXW410x: EN60950-1 GXW4108: UL60950-1 (in addition)

CONFIGURATION GUIDE

CONFIGURATION WITH WEB BROWSER

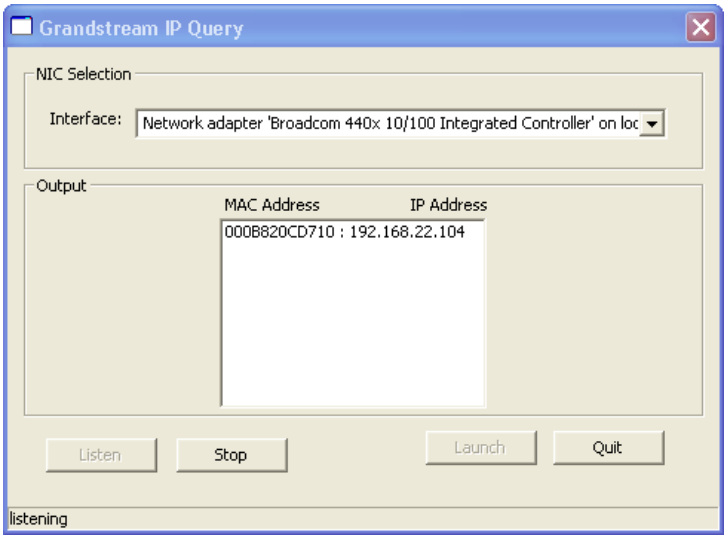
The GXW410x gateways have an embedded Web server that will allow a user to configure the IP phone through any web browser.

Examples of GUI interfaces can be downloaded @
http://www.grandstream.com/support/gxw_series/gxw410x/documents/gxw410x_gui.zip

ACCESSING THE WEB CONFIGURATION MENU

1. Navigate your browser to: http://www.grandstream.com/DOWNLOAD/IPQuery/IPQuery.zip
2. Run the Grandstream IPQuery tool, that you just downloaded.
3. Click on

button in order to begin device detection
4. The detected devices will appear in the Output field



END-USER CONFIGURATION

When the Web Configuration Menu is accessed, the GXW410x will respond with a login screen. There are two default passwords for the login page:

User Level:	Password:	Web pages allowed:
End-user Level	123	Only Status and Basic Settings
Administrator Level	admin	All pages can be browsed.

After login, the following page is the **Basic Configuration** page, explained in detail in **Table 6: Web Log-in Definition**.

TABLE 6: WEB LOG-IN DEFINITIONS (BASIC SETTINGS PAGE)

Web Access	Select HTTP or secure HTTPS protocol for Web Access
Web Port	By default, HTTP uses port 80 and HTTPS uses port 443. This field is for customizable web port.
End-user Password	This contains the password to access the End-user Web Configuration Menu (Status and Basic Settings). This field is case sensitive with a maximum length of 25 characters.
IP Address	There are two modes to operate the GXW410x: **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXW410x acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The GXW410x will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields.
DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP domain	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
DHCP vendor class ID	Used by clients and servers to exchange vendor-specific information. Default is Grandstream GXW410x.
PPPoE account ID	PPPoE user name. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
Preferred DNS server	This field will let the user enter a preferred DNS server to be used instead of the one acquired by the service provider.
Time Zone	Controls how the date/time is displayed according to the specified time zone.
Allow DHCP Option 2 to override Time Zone Settings	Default is No. If set to Yes, time zone settings will originate from the DHCP server.

Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to "Yes" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time. The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving Both start-time and end-time have the same syntax: Month ,day ,weekday ,hour ,minute month: 1,2,3,..,12 (for Jan, Feb, .., Dec) day: [+|-]1,2,3,..,31 weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. hour: hour (0-23), minute: minute (0-59) If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" us not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value for "Automatic Daylight Saving Time Rule" shall be set to "03,11,0,02,00;11,04,0,02,00;60" which is the rule for US. Examples: US where daylight saving time is applicable: 03,11,0,02,00;11,4,0,02,00;60 This means the daylight saving time starts from 11th March at 2AM and ends November 4th at 2AM. The saving is 60 minutes (1hour).

You may also access the Device Status page which provides details of the GXW product. The Device Status page terms are defined in **Table 7: Status Page Definitions**.

Table 7: Status Page Definitions

Hardware Revision	Hardware version number: Main Board, Interface Board
MAC Address	The device ID in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows WAN IP address of GXW410x
Product Model	This field contains the product model info (GXW4104 or GXW4108)
Software Version	Program: This is the main software release. Boot and Loader are not changed often.
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the different SIP Accounts configured under Channels page are successfully registered to the SIP server(s).

FXO Line Connected	This field will give the status of each physical FXO Line connected to the Gateway. It will update the status regularly. **Yes** - Connected and Idle **Busy** - Connected and Busy **No** - Not connected Additionally it will also provide real time Caller ID information of Incoming as well as Outgoing calls.
PPPoE Link Up	This field shows whether the PPPoE connection is running if connected to DSL modem.
Detected NAT Type	This field shows what kind NAT the GXW410x is connected to via its WAN port. It is based on STUN protocol.

Advanced User Settings

Advanced User Configuration

The end-user needs to login to the advanced user configuration page the same way as for the basic configuration page.

Advanced User configuration includes the end-user configuration and advanced configurations including: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

Table 8: Advanced Configuration Page Definitions

Admin Password	Administrator password. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reasons. The maximum password length is 25 characters.
G723 Rate	G723 encoding rate (6.3kbps or 5.3kbps)
Voice Frames per Tx	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. For example if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the gateway will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48. Its range lies from 0 to 63.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. 802.1q / VLAN tag: Default value is 0. Range lies from 0 to 4095. 802.1p Priority value: Default value is 0. Range lies from 0 to 7. *** The above 2 settings need to be supported on the network and then configured accordingly on the GXW410x. Incorrect configuration will cause blocked access, which will result in Factory Reset as the only option to renew access.
Local RTP port	This parameter defines the local RTP-RTCP port pair the GXW410x will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP and so on. The default value is 5004.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXW410xs are behind the same NAT.
Keep-alive interval	This parameter specifies how often the GXW410x sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.

STUN Server	IP address or Domain name of the STUN (Simple Traversal of UDP through NATs) server.
Firmware Upgrade & Provisioning	This radio button will enable GXW410x to download firmware or configuration file through either TFTP or HTTP.
Via TFTP Server	If selected, the GXW410x will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 5 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 25 or 30 minutes.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/ Here ":6688" is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, GXW410x will only do HTTP download once at boot up.
Firmware Server Path	IP address or domain name of firmware server.
Config Server Path	IP address or domain name of configuration server.
Firmware File Prefix	Default is blank. If configured, GXW400X will request firmware file with the prefix. This setting is useful for ITSPs. End-user should keep it blank.
Firmware File Postfix	Default is blank. End-user should keep it blank.
Config File Prefix	Default is blank. End-user should keep it blank.
Config File Postfix	Default is blank. End-user should keep it blank.
Allow DHCP Option 66 to override server	Default value is No. If set to Yes, configuration file will originate from the DHCP server.
Automatic Upgrade	Choose ***Yes*** to enable automatic upgrade and provisioning. In "Check for new firmware every" field, enter the number of minutes to enable GXW410x to check the server for firmware upgrade or configuration. When set to **No**, GXW410x will only do upgrade once at boot up. Other options are: " Always check for New Firmware." " Check New Firmware only when F/W pre/suffix changes" " Always skip the Firmware check"
Authenticate Conf File	If set to Yes, configuration file is authenticated before acceptance. This protects the configuration from an unauthorized change.

DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833. Default is 101.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.
Allow DHCP Option 42 to override an NTP server	Default value is No. If set to Yes, the NTP server will originate from the DHCP server.
Enable Video Surveillance	When set to Yes, GXW410x will start converting video feed received from analog camera to IP packets. In order to view this video feed, please follow instructions given under Video Surveillance chapter on page 23.
RTSP Port	By default it is 554.

Configuring the FXO Channels

Configuring the FXO channels on the GXW410x gateway is an easy process. Follow the GUI interfaces. The Device Status page terms are defined in **Table 9: FXO Lines Configuration Definitions.** An example of the Channel Dialing Configuration is shown in Figure 6. Please note the default is always configured. The user has the option to change the default settings as described in the Table 9.

Table 9: FXO Lines Configuration Definitions

	Syntax for Channel Specific Settings: Default: ch1-8:X; {all channels 1 to 8 are set to use value X} Additional Examples: ch1,3-6:10;ch2,7-8:12 {channels 1,3,4,5 and 6 are set to use value 10 while channels 2,7 and 8 are set to use value 12.
Enable Current Disconnect	When set to Y, Current Disconnect is enabled. Certain PSTN COs require this to be enabled, in order to realize disconnect signal from PSTN side. Default is Y. **If enabled use threshold**: Default is 100ms. Range is 40ms to 800ms. Certain PSTN service providers have a threshold time within which the line stabilizes after off-hook. It is entirely dependent on provider, however if you experience PSTN line detection issues, please modify this setting appropriately in 100ms increments. If you are not sure if this option should be enabled please refer to Table 10 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper PSTN configuration that the gateway should have to work with your Service Provider or analog PBX.
Enable Tone Disconnect	Default is No. If PSTN provider uses call progress tones then it should be set to Yes in order to realize disconnect tone. Please configure accurate Call Progress Tones on Channels web page based on PSTN provider (or traditional PBX) settings. If you are not sure if this option should be enabled or what Call Progress Tones are required please refer to Table 10 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper PSTN configuration that the gateway should have to work with your Service Provider or analog PBX.
Enable Polarity Reversal	Default is No. This should be set to Yes only if the FXO lines are subscribed to PR service from PSTN Service provider. It is merely a PR detect feature. ***Note: If there is no PR service from provider on the FXO line, and this setting is configured to Yes, calls will not be successful.
Silence Timeout	Terminate call after long silence detected. Default is 60 seconds, max 65536.
AC Termination Impedance	Selects the impedance of the analog line connected to the FXO port on the GXW410x. Here is some basic information which may be helpful for initial configuration: 600 Ohm – North America; 270 Ohm + (750 Ohm || 150 nF) -- Most of Europe 220 Ohm + (820 Ohm || 120 nF) – Australia, New Zealand 220 Ohm + (820 Ohm || 115 nF) – Austria, Bulgaria, Germany, Slovakia, South Africa 370 Ohm + (620 Ohm || 310 nF) – UK., India If this parameter is not configured properly you may experience echo or noise in the line. Please refer to Table 10 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the correct impedance value to match your lines.

Wait for Dial-tone	Default is No. When set to Yes, the gateway will recognize dial-tone from the Central Office (CO) before it completes a call. If you can't make an outbound call, set this is **Yes**. if this is set to Yes make sure you have configured the dial tone settings correctly in the Channels tab and that there is not any major noise interference in the line.
Stage Method	Syntax - ch1-8:1; {all channels 1 to 8 are set to value 1 or 2} Stage method can be set to either 1 or 2. Set this parameter to 1 if you need to make a direct PSTN call from a VOIP endpoint. When you set it to 2, you will first dial one of the VOIP channel accounts from the VOIP endpoint, this will result in getting a PSTN line dial-tone to then dial out the destination PSTN number. Most implementations require this setting to be configured to 1.
Min. Delay before Dial PSTN	Default is 500ms. This needs to be equal to or greater than the Current Disconnect threshold setting. Once the threshold is reached the gateway can dial out. This parameter should only be used if there are PSTN line detection issues.
Unconditional Call Forward to VOIP:	This is an extremely important setting to make sure incoming PSTN calls are picked up and forwarded to the correct VOIP destination. **User ID** - This parameter allows users to configure a User ID or extension number to be automatically dialed upon FXO line off-hook. **SIP Server** - You also need to specify the Profile of the user id configured above (p1 stands for Profile 1, p2 stands for Profile 2 and so on). **SIP Destination Port** - Along with the user-id and Profile, you also have the option to choose the destination port where you would like to send the call. By default it should be set to ch1-x:5060; (x can be 4 or 8 depending on number of ports). We can also specify a different destination for each port. For example under User ID we can type in: ch1:104;ch2:227;ch3-5:501;ch6,7:856. Under Sip Server we can type in: ch1:p1;ch2-4:p2;ch5:p3 Under Sip Destination Port we can type in: ch1-2:5060;ch2:7080;ch3-8:5066++
Number of Rings Before Pickup	Default is 4. This is the number of rings the gateway will wait to send the call to the VOIP side in case the Caller ID has yet to be detected. If there's CID information the call will be sent right away. If your lines don't have the CID service set this to 1.

Caller ID Scheme	The GXW410x supports 5 different types of schemes: 1. Bellcore (US standard) 2. ETSI-FSK during ringing 3. ETSI-FSK prior to ringing with DTAS 4. ETSI-FSK prior to ringing with LR 5. ETSI-FSK prior to ringing with PR 6. ETSI-DTMF during ringing 7. ETSI-DTMF prior to ringing with DTAS 8. ETSI-DTMF prior to ringing with PR 9. ETSI-DTMF prior to ringing with PR 10. SIN 227 - BT 11. NTT (Japanese standard) Please check with your PSTN service provider (or traditional PBX specs) for which caller ID scheme they/it support. If you are not sure about which to use please refer to Table 10 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper Caller ID Scheme so the gateway can properly detect the Caller ID. Similarly to the cases explained above we can specify a caller ID scheme for each channel independently.
Caller ID Transport type	Default is “relay via From header”. You may also select : “relay via P_Asserted_Identity header” “Disable” : Caller ID feature will be disabled. “Send anonymous” : All calls forwarded to VOIP end will be sent as anonymous.
T.38 Setting	This setting allows you to make several options related to facsimile. You can select the method: T.38 or Pass through (G711) You can select the fax transmission rates (2400/4800/7200/9600/12000/14400bps) You can enable or disable ECM (Error Checking Mode)

TABLE 10: FXO LINE TEST TAB DEFINITIONS

	Note: The user can only test the parameters for only one of the PSTN lines at the same time. In all cases please enter the telephone numbers as if the lines were to dial each other locally. For the AC Impedance Test we only need to select the line to be tested by clicking on the AC impedance box corresponding to that line, telephone numbers are optional. Remember that the AC impedance test is usually used to reduce the echo that might be present in the line. For the CPT test (call progress tones) we will test current disconnect as well. You will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test. For CID detection you will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test. To perform the test, please select the line you want to test and the desired test to be performed. Enter the information for this line as well as a second line if necessary. Then click on the update button and then reboot. Log back in and now you should see the information for the line selected as well as the check box already marked already there. Go ahead and start the test now, please wait a few minutes until the test is done. Notes: It is not required to enter a telephone number when testing for impedance, as the system does not place any actual calls for the test. If you log into the Web Interface while the test is running will not interrupt the process.
Line #	Enter the telephone (PSTN) number that corresponds to this line. Enter it as if you were going to dial it locally
AC Impedance	Select this box if you want to test for impedance on the line that is on the same row as the checked box. Remember that you can only check one item at the same time.
CPT Detection	Select this box if you want to test for call progress tones and current disconnect threshold on the line that is on the same row as the checked box. Remember that you can only check one item at the same time.
External Number	Enter an external telephone (PSTN) number to be used as an auxiliary number for the test. This is used only if we do not have at least 2 PSTN lines connected to the gateway. This is only used for CPT and current disconnect threshold testing. In order to use this function you will have to monitor the Syslog output. This is only reserved for very advanced users
External Call Timeout	This is the time the GXW will wait for the external telephone number to pick up during the test.

Apply test results automatically	Default is No. If selected on Yes, then all the results from the test will be applied automatically. If you select No you will have to monitor the Syslog output. This is only reserved for very advanced users.
Apply test results to all ports	Default is No. If selected on Yes, then all the test results will be applied to all ports on the gateway. If all the lines belong to the same service provider or PBX it will make sense to apply the results to all ports.
Error Timeout	This is the time the gateway will wait to exit the test mode, when something unexpected or an error has occurred.

TABLE 11: CHANNELS PAGE DEFINITIONS

	Note – The channels here are basically SIP endpoints that will act as clients registering to the SIP Server configured under the appropriate Profile page.
Channels	It should be set same as the channel number (i.e 1, 2..4 or 8 depending on number of FXO ports). *It is NOT the same as SIP Account ID.*
SIP User ID	This is the SIP account information. Enter the SIP User ID part of the account.
Authentication ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to, or different from SIP User ID.
Authentication Password	SIP account password needs to be entered here. Note: After entering the password, it will show up as blank but the password still remains active.
Profile ID	Select the corresponding Profile ID (1/2/3). Profiles are SIP Server configurations.
Call Progress Tones	Using these settings, user can configure tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **ON** is the period of ringing (ON time in ms) while **OFF** is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. • "Dial tone" • "Ringback tone" • "Busy/Re-order tone" • "Confirmation tone" Please refer the document below to determine your local call progress tones **(http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf)** or run the FXO Line Test (Table 10).
Channel Voice Settings	Channel voice settings mentioned below.
Tx to PSTN Audio Gain (dB)	Allows user to set a value in dB for transmission to PSTN Audio Gain. Default is 1. Range is from -12 to 12dB.
Rx from PSTN Audio Gain (dB)	Allows user to set a value in dB for receive from PSTN Audio Gain. Default is 0. Range is from -12 to 12dB.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.

Echo Cancellation	When set to Y, Echo cancellation is enabled.
Channel specific Setting	Channel specific settings mentioned below.
DTMF Method	This parameter specifies the mechanism to transmit DTMF digits. There are7 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low bit-rate codec), via RTP (RFC2833), or via SIP INFO. Multiple DTMF transmission schemes can be selected. • 1 – in-audio • 2 – RFC2833 • 3 – in-audio and RFC2833 • 4 – SIP Info • 5 – in-audio and RFC2833 • 6 – SIP Info and RFC2833 1. 7 – in-audio, RFC2833, and SIP Info
No Key Entry Timeout	Default is 4 seconds.
Local SIP Listen Port	Default is ch1-8:5060++;. The ++ indicates increments by 2, so port 1 is set at 5060, port at 5062 and so on. This setting can be used with Round Robin and/or Flexible setting below to configure different ports to be placed under different Round Robin groups.
SRTP Mode	Default is disabled for all ports. The user can select to either enable it but not force it or force it on an individual port basis. When used the communication will be sent using Secure RTP.
Round Robin and/or Flexible	Default is rr:1-8; The syntax is pretty straight-forward here. The rr stands for Round Robin and the numbers stand for the ports that belong to that round robin group. **For example:** rr:1-8; -> Round robin within the first 8 ports i.e. outgoing calls will be forwarded to the next available port within the group of ports 1 to 8. rr:1,3-6,8;rr:2,7; -> Round robin within port 1,3,4,5,6 and 8; Second round robin group within ports 2 and 7 i.e. outgoing calls to ports 1,3,4,5,6 and 8 will be forwarded to the next available port within this group ONLY. Outgoing calls to port 2 and 7 will be forwarded to the next available port between ports 2 and 7 ONLY. *** In order to terminate a call on FXO port 2 or 7 you will need to change its Local SIP Listen port accordingly.*
Prefix to specify Port (1 stage dialing method)	Default is 99. Syntax to USE this feature: prefix# (that is 99) + ch# (could be anything from 1 to 8) + dialing# will result in this call forwarded to FXO port (ch#) immediately.

Table 12: Channels Page Definitions

Dial Plan	The Dial Plan feature implemented is applicable for VOIP to PSTN calls only. You may configure a dial plan based on the following grammar: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0,*,#,A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; xx+ - at least 2 digit number; xx. - at least 2 digit number; ^ - exclude; [3-5] - any digit of 3, 4, or 5; [147] - any digit 1, 4, or 7; <2=011> - replace digit 2 with 011 when dialing WARNING - illegal input will fall back to default **Example 1**: {[369]11 | 1617xxxxxxx} - Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617. **Example 2**: {^1900x+ | <=1617>xxxxxxx} - Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers. **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} - Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing Default: PSTN Outgoing - {x+} **Note**: If you do not plan to use this feature set to default {x+}
Hook flash Duration (X10ms)	Default 600ms. This value can accept any value in the 100-2000ms range.
Use DTMF Parameter from RFC2833 or SIP Info	Default Yes, No means to use DTMF parameter settings according to DTMF Digit Length, DTMF Digit Volume and DTMF Dial Pause.
DTMF Digit Length	Default value is 100ms. Please not that the value will be multiplied by 10ms
DTMF Digit Volume	Default value is -11dB.
DTMF Dial Pause	Default value is 100ms. Please note that the value will be multiplied by 10ms.

Profiles

Profiles are basically IP-PBX / SIP Server configuration templates. If you have more than one IP-PBX system or SIP Server that you would like to use with the GXW410x, then you can configure Profile 2 or 3. **Note:** Make sure you select the correct profile for each channel under Channels PAGE

Table 13: Profile Page Definitions

Activate Profile	When set to Yes the SIP Profile is activated.
Profile Name	A name to identify a Profile.

SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW410x for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV to look up server.
User ID is Phone Number	If the GXW410x has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter controls whether the GXW410x needs to send REGISTER messages to the SIP Server. The default setting is "Yes".
Unregister on Reboot	Default is No. If set to yes, the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) for the GXW410x to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
SIP Registration Failure Retry Wait Time	This parameter is most used by Service Providers. It prevents message REGISTER overload of SIP Server in case of downtime due to maintenance or power failure. By increasing interval length, common message load is decreased. Interval range is 1 – 3600 seconds.
NAT Traversal	This parameter defines whether the GXW410x NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW410x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXW410x will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXW410x will attempt to use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the GXW410x will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Early Dial	Default is No. Use only if proxy supports 484 response.
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Request Timer	If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.

Force Timer	If selecting “Yes” the phone will use session timer even if the remote party does not support this feature. Selecting “No” will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
Enable 100rel	The use of the PRACK (Provisional Acknowledgement) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN inter-networking is to be supported. A user’s request to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
Refer-to uses Target Contact	Default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
INVITE Ring-no-answer Timeout	In case incoming call has arrived from PSTN to VoIP and INVITE message was generated by GXW device, the call will be disconnected after pre-configured timeout if not answered by VoIP extension.
Preferred Vocoder	The GXW410x supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 8”.
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

Saving the Configuration Changes

When changes are made, press the “Update” button in the Configuration Menu. The GXW410x will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the GXW410x after all changes are made.

Rebooting from Remote

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The following screen will indicate that rebooting is underway.

The user can re-login to the unit after waiting for about 30 seconds.

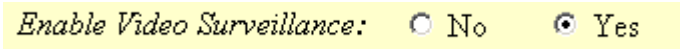
VIDEO SURVEILLANCE

The GXW410x gateway can be used with an Analog Surveillance CCD Camera to perform video surveillance function. This application should be used in a LAN environment or when both sides have public IP address.

VIDEO SURVEILLANCE PROCEDURES

- **Gateway side:**

  1. In the ADVANCED SETTING page, find the following field and change from default setting NO to YES, reboot the device.

  2. Connect an analog surveillance camera to the VIDEOIN connection at the back panel of the unit.

- **PC side** (Monitor Device):

  1. Download VLC from http://www.videolan.org/vlc/. This is the only player so far that supports RFC 3984.
  2. Launch VLC.
  3. Go to Preferences->Input/Codecs->Demuxers->H264, check "Advanced options" in the bottom. The option "Frames per Second" will show. Change that value to 5 and then save.
  4. Go to Preferences->Input/Codecs->Access modules->Real RTSP, check "Advanced options" in the bottom. The option "Caching value (ms) will show. Change that value to 1000 and then save. You may change it to a smaller value to reduce the delay.
  5. If the viewer is under NAT, go to Preferences->Demuxers->Access modules->RTP/RTSP, check "Advanced options" in the bottom. The option "Use RTP over RTSP (TCP)" will show. Check that option box. *(Grandstream does NOT recommend this network environment)*
  6. Close the Preferences window and go to File->Open Network Stream:

  a) Select RTSP as the protocol

  b) Enter the URL in the format of rtsp://admin:*ADMIN_PASSWORD@DEVICE_IP_ADDRESS:DEVICE_RTSP_PORT*.
  Change the blue text according to your configuration:
  - **ADMIN_PASSWORD** is the device's web configuration password for admin.
  - **DEVICE_IP_ADDRESS** is the device IP.
  - **DEVICE_RTSP_PORT** is the RTSP port setting of the device.

  If the port uses default value 554, the port portion can be omitted from the URL

  c) Click OK to start the video.

FIGURE 7: SCREEN-SHOT OF VIDEO SURVEILLANCE*

* PC client side running VLC as monitoring station

FIRMWARE UPGRADE

Our latest official release can be downloaded from: http://www.grandstream.com/firmware.html.

Firmware (or software) upgrades can be done either via TFTP or HTTP. The corresponding configuration settings are on the configuration page. End-users should NOT modify the configuration settings that are useful for ITSPs. To upgrade your unit firmware, follow these steps:

1. Under Advanced Settings web page, enter your TFTP or HTTP Server IP address (or FQDN) next to the "Firmware Upgrade: Upgrade Server" field.
2. Select via TFTP or HTTP accordingly.
3. If you plan to use Automatic Upgrade, set it to "Yes", otherwise No (this will make it check for upgrade every time you reboot).

UPGRADE THROUGH HTTP

To upgrade firmware via HTTP, the field *"Firmware Upgrade and Provisioning: Upgrade Via"* needs to be set to HTTP. The *"Firmware Server Path"* should be set to where the firmware files are located.

For example: The user can use the following URL in the Firmware Server Path: **firmware.mycompany.com: 6688/Grandstream/1.3.1.6** where firmware.mycompany.com is the FQDN of the HTTP server. It can also be in IP address format. ":6688" is the TCP port the HTTP server listening to, default http server listens to port 80. "/Grandstream/1.3.1.6 is the RELATIVE directory to the root dir on HTTP web server.

UPGRADE THROUGH TFTP

To upgrade firmware via TFTP, set the field *"Firmware Upgrade and Provisioning: Upgrade Via"* to TFTP. The TFTP server can be configured in either IP address format or FQDN.

To configure the TFTP server via the Web configuration interface, follow these five steps:

1. Open your browser to input the IP address of the GXW410x.
2. Enter the admin password to enter the configuration screen.
3. Enter the TFTP server address or URL in the "Firmware Server Path" field near the bottom of the configuration screen.
4. Once the "Firmware Server Path" is set, update the change by clicking the "Update" button.
5. Reboot or power cycle the unit.

If the configured upgrade server is found and a new code image is available, the GXW410x will retrieve the new image files by downloading them into the GXW410 gateway's SRAM. During this stage, the GXW410x gateway's LED will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW410x gateway will stop the TFTP process and simply boot using the existing code image in the flash.

Firmware upgrade process may take as long as 20 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Grandstream recommends conducting firmware upgrades in a controlled LAN environment if possible.

DOWNLOAD TFTP SERVER

For users who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Services section of Grandstream's Web site to obtain this TFTP server IP address. Alternatively, user can download and install a free TFTP or HTTP server in his LAN for a firmware upgrade.

A free Windows version TFTP server can be downloaded from:
http://support.solarwinds.net/updates/New-customerFree.cfm.

Directions for Downloading TFTP Server:

1. Unzip the file and put all of the files under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW410x in the same LAN segment.
3. Go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page.
5. Configure the Firmware Server Path with the IP address of the PC.
6. Update the change and reboot the unit.

You can also download the free HTTP server from http://httpd.apache.org or just use Microsoft IIS web.

RESTORE FACTORY DEFAULT SETTING

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

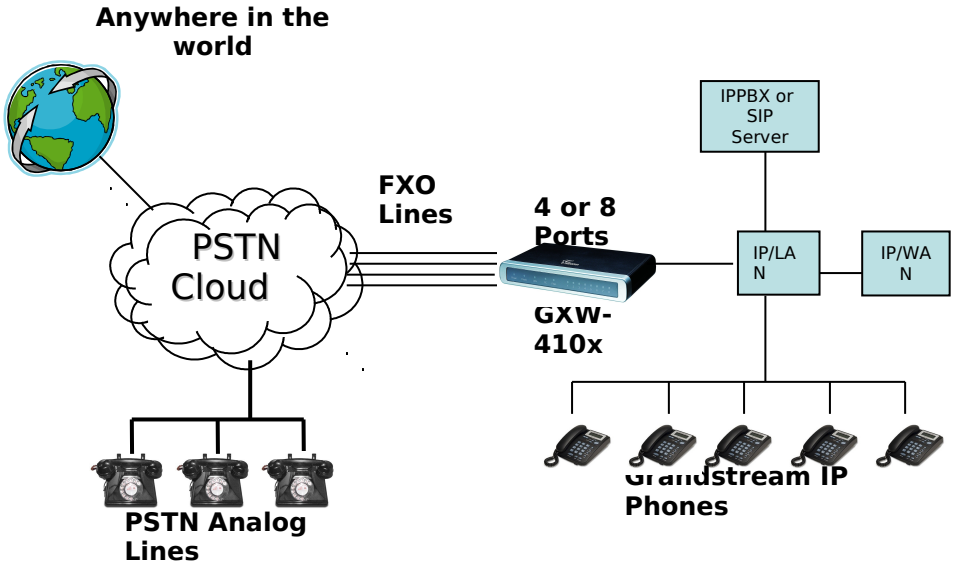
The ONLY way to restore default factory settings is as follows:

1. Unplug the Ethernet cable.
2. Locate the needle sized hole on the back panel of the gateway next to the Power connection.
3. Enter a thin object in this hole and keep it pressed for about 7 seconds.
4. You will see the Network port LEDs (green and orange) go off and on simultaneously; this indicates the reset went through.
5. All settings have been erased and the gateway is back to factory settings.

Examples of GXW410x Configurations

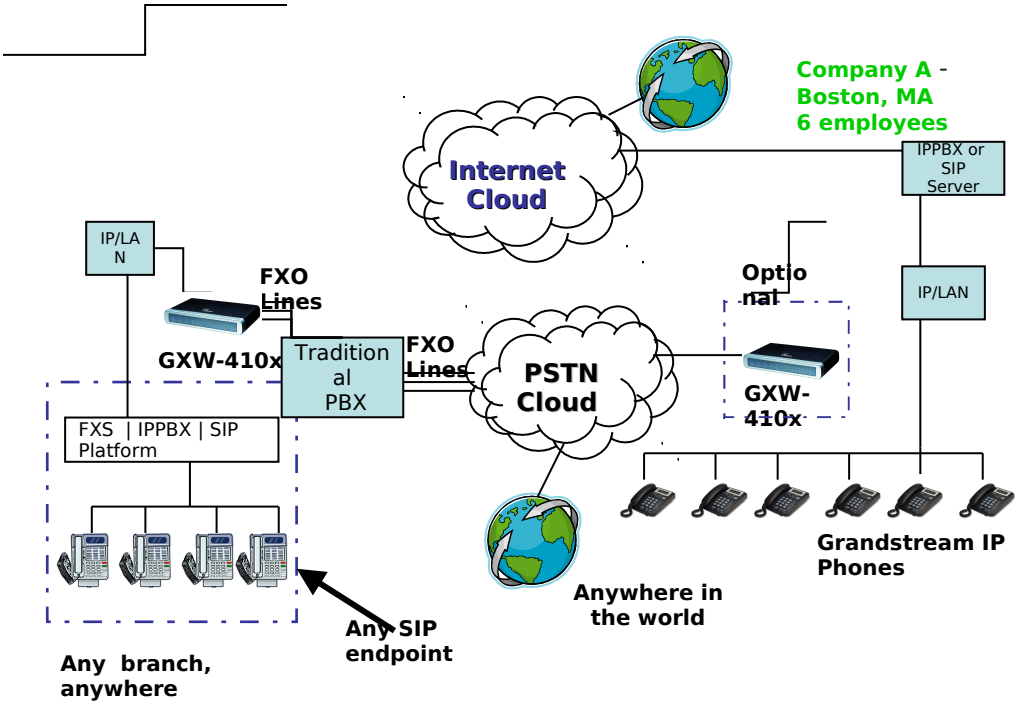
Application One: GXW connected with an IP-PBX or SIP Server

Scenario: A business with a traditional phone system (with or without broadband access) and an IP PBX or SIP Servers connecting to an Internet Telephone Service Provider (ITSP).



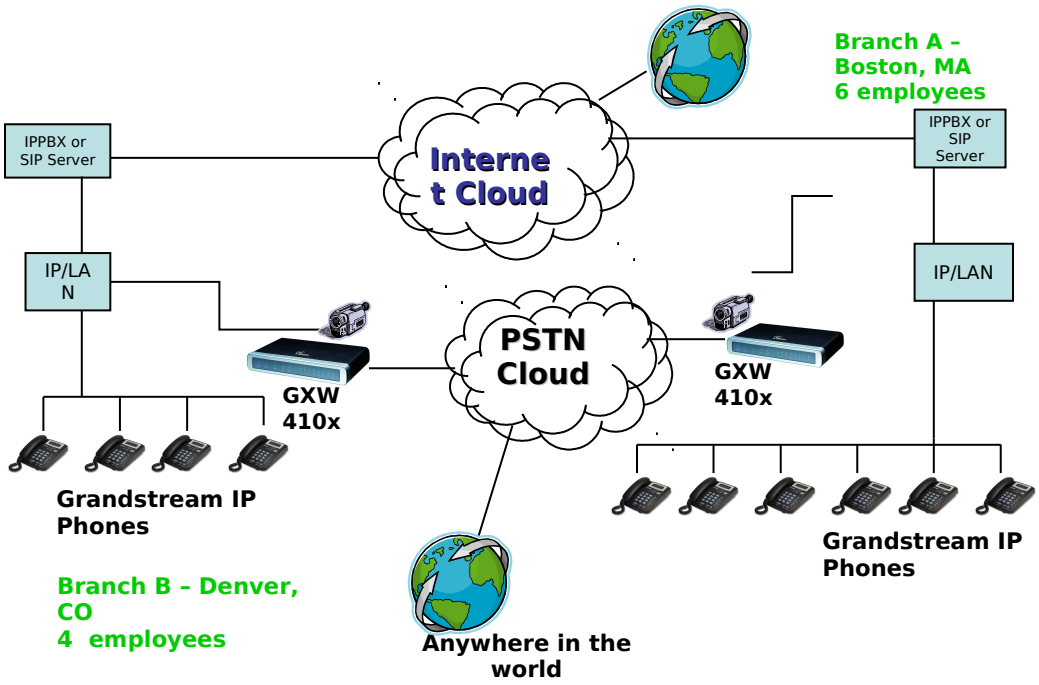
Application Two: GXW to Extend a Traditional PBX Scenario

Scenario: a small business with traditional analog PBX lines and broadband access who want to extend their traditional PBX to virtually anywhere in the world, using the Internet. (Any SIP End point, such as Grandstream BugeTone, HandyTone, GXP-2000 or GXV-3000 are needed in this scenario)



Application Three: GXW connected with an IP-PBX or SIP Server and Video Surveillance

Scenario: The GXW410x offers an additional video surveillance port which can be configured for surveillance. It is the only small business analog gateway that offers this security feature.



Application Four: Using a GXW for Pure IP- IP Communication Configuration

Scenario Four: The GXW410x offers an IP to IP pure IP Communications System configuration, where all locations use IP phones.